Grandstream Networks, Inc.

GXP1450 SIP Enterprise Phone

TABLE OF CONTENTS
GXP1450 USER MANUAL

WELCOME **3**

INSTALLATION **4**

EQUIPMENT PACKAGING 4
CONNECTING YOUR PHONE 4
SAFETY COMPLIANCES 4
WARRANTY 4

PRODUCT OVERVIEW **5**

USING THE GXP1450 SIP ENTERPRISE PHONE **8**

GETTING FAMILIAR WITH THE LCD 8
MAKING PHONE CALLS 11
ANSWERING PHONE CALLS 14
PHONE FUNCTIONS DURING A PHONE CALL 14
CALL FEATURES 17
CUSTOMIZED LCD SCREEN & XML 18

CONFIGURATION GUIDE **19**

CONFIGURATION VIA KEYPAD 19
CONFIGURATION VIA WEB BROWSER 22
SAVING THE CONFIGURATION CHANGES 39
REBOOTING THE PHONE REMOTELY 39

SOFTWARE UPGRADE & CUSTOMIZATION **40**

FIRMWARE UPGRADE THROUGH TFTP/HTTP 40
CONFIGURATION FILE DOWNLOAD 41

RESTORE FACTORY DEFAULT SETTING **42**

TABLE OF FIGURES
GXP1450 USER MANUAL

Figure 1: GXP1450 Keypad Layout 10
Figure 2: Keypad GUI Flow 20

TABLE OF TABLES
GXP1450 USER MANUAL

Table 1: Equipment Packaging 4
Table 2: GXP1450 Connectors 4
Table 3: GXP1450 Feature Guide 5
Table 4: GXP1450 Key Features in a Glance 5
Table 5: GXP1450 Hardware Specifications 5
Table 6: GXP1450 Technical Specifications 6
Table 7: LCD Display Definition 8
Table 8: LCD Icons 8

Table 10: GXP1450 Call Features 17
Table 11: Key Pad Configuration Menu 19
Table 12: Device Configuration - Status 23
Table 13: Device Configuration – Settings/Basic Settings 23
Table 14: Device Configuration – Settings /Advanced Settings 26
Table 15: SIP Account Settings 32

GUI INTERFACE EXAMPLES

GXP1450 USER MANUAL

http://www.grandstream.com/products/gxp_series/general/documents/gxp21xx_gui.zip

1. Screenshot of Configuration Login Page
2. Screenshot of Status Page
3. Screenshot of Basic Setting Configuration Page
4. Screenshot of Advanced User Configuration Page
5. Screenshot of SIP Account Configuration Page
6. Screenshot of Saved Configuration Changes Page
7. Screenshot of Reboot Page

Welcome

GXP1450 is a next generation enterprise grade IP phone that features 2 lines with 2 SIP accounts, a 180x60 backlit graphical LCD, 3 XML programmable context-sensitive soft keys, dual network ports with integrated PoE and 3-way conference.

The GXP1450 delivers superior HD audio quality, rich and leading edge telephony features, personalized information and customizable application service, automated provisioning for easy deployment, advanced security protection for privacy, and broad interoperability with most 3rd party SIP devices and leading SIP/NGN/IMS platforms. It is a perfect choice for enterprise users looking for a high quality, feature rich IP phone with affordable cost.

Caution: *Changes or modifications to this product not expressly approved by Grandstream, or operation of this product in any way other than as detailed by this User Manual, could void your manufacturer warranty*.

Warning: *Please do not use a different power adaptor with the GXP1450 as it may cause damage to the products and void the manufacturer warranty.*

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Installation

EQUIPMENT PACKAGING

Table 1: Equipment Packaging

	GXP1450
Main Case	Yes
Handset	Yes
Phone Cord	Yes
Power Adaptor	Yes
Ethernet Cable	Yes
Base Stand	Yes
Quick Start Guide	Yes

CONNECTING YOUR PHONE

The connectors of the GXP1450 are located on the bottom of the device.

Table 2: GXP1450 Connectors

PC	10/100Mbps RJ-45 ports for PC (downlink) connection
LAN	10/100Mbps RJ-45 port for LAN (uplink) connection. Supports PoE (802.3af)
Power Jack	5V DC power port; UL Certified
Headset Jack	RJ9
Handset Jack	RJ9

SAFETY COMPLIANCES

The GXP1450 complies with FCC/CE and various safety standards. The GXP1450 power adaptor is compliant with the UL standard. Only use the universal power adaptor provided with the GXP1450 package. The manufacturer's warranty does not cover damages to the phone caused by unsupported power adaptors.

WARRANTY

If you purchased your GXP1450 from a reseller, please contact the company where you purchased your phone for replacement, repair or refund. If you purchased the product directly from Grandstream, contact your Grandstream Sales and Service Representative for a RMA (Return Materials Authorization) number before you return the product. Grandstream reserves the right to remedy warranty policy without prior notification.

Product Overview

Table 3: GXP1450 Feature Guide

Features	GXP1450
LCD Display	180x60 pixel
Number of Lines	2
Programmable Soft Keys	3
Extension Module	N/A

Table 4: GXP1450 Key Features in a Glance

Features	Benefits
Open Standards Compatible	SIP RFC3261, TCP/IP/UDP, RTP, HTTP/HTTPS, ARP/RARP, ICMP, DNS (A record, SRV and NAPTR), DHCP, PPPoE, TELNET, TFTP, NTP, STUN, SIMPLE, TR-069, 802.1x, and etc
Superb Audio Quality	Advanced Digital Signal Processing (DSP), Silence Suppression, VAD, CNG, AGC
Network Interfaces	Dual 10/100mbps Ethernet ports with integrated PoE
Feature Rich	Traditional voice features including caller ID, call waiting, hold, transfer, forward, block, auto-dial, off-hook dial
Advanced Features	2 line keys with dual-color LED and 2 SIP accounts, 3 way conferencing, backlit graphic 180x60 LCD, 3 XML programmable context sensitive soft keys, 5 navigation keys, 10 dedicated buttons for HOLD, TRANSFER, CONFERENCE, VOLUME, HEADSET, MUTE, SPEAKERPHONE, SEND/REDIAL, PHONEBOOK, MESSAGE
Advanced Functionality	Customized downloadable ring-tones, SRTP, SIP over TLS, multi-language support and XML enabled, adjustable positioning angles, wall mountable, AES encryption, automatic multimedia service (eg., weather information)

Table 5: GXP1450 Hardware Specifications

	GXP1450
LAN Interface (Ethernet ports)	Two (2) 10/100 Mbps Full/Half Duplex Ethernet Switch with LAN and PC port with auto detection
Graphic LCD Display	180x60 pixel
Expansion Module Support	N/A
Headset Jack	RJ9
Call Appearance LED	2 Dual color (green/red)
Power over Ethernet	Built-in auto-sensing: Cisco and IEEE 802.3af standard
Universal Switching	Input: 100-240VAC 50-60 Hz
Power Adaptor	Output: +5VDC, 800mA, **UL** certified

Dimension	186mm (W) x 210mm (L) x 81mm (D)
Weight	0.8KG
Temperature	32 –104° F/ 0 – 40° C
Humidity	10% – 90% (non-condensing)
Compliance	FCC Part 15 (CFR 47) Class B; EN55022 Class B, EN55024, EN61000-3-2, EN61000-3-3, EN 60950-1; AS/NZS CISPR 22 Class B, AS/NZS CISPR 24, RoHS; UL 60950 (power adapter)

Table 6: GXP1450 Technical Specifications

Lines	2 lines with 2 independent SIP accounts
Protocol Support	Support SIP 2.0, TCP/UDP/IP, PPPoE, RTP, SRTP by SDES, HTTP, ARP/RARP, ICMP, DNS, DHCP, NTP, TFTP, SIMPLE, TR-069, 802.1x Support 2 SIP accounts Support SIP Dialog package (RFC 4235) Support for SIP MESSAGE method (RFC 3428)
Display	Backlit graphic LCD display, up to 4 level grayscale
Feature Keys	HOLD, TRANSFER, CONF, VOLUME, HEADSET, MUTE, SPEAKERPHONE, SEND/REDIAL, PHONEBOOK, MESSAGE, 3 XML Programmable Softkeys, 5 Navigation keys,
Device Management	NAT-friendly remote software upgrade (via TFTP/HTTP) for deployed devices including behind firewall/NAT Auto/manual provisioning system, Web GUI Interface Address Book
Audio Features	Full-duplex hands-free speakerphone Advanced Digital Signal Processing (DSP) Dynamic negotiation of codec and voice payload length Support for G.723,1 (5.3/6.3K), G.729A/B, G.711 a/μ-law, G.726-32, G.722 (wide-band), and iLBC codecs In-band and out-of-band DTMF (In audio, RFC2833, SIP INFO) Silence Suppression, VAD (voice activity detection), CNG (comfort noise generation), ANG (automatic gain control) Acoustic Echo Cancellation (AEC) with Acoustic Gain Control (AGC) for speakerphone mode, Support side tone Adaptive jitter buffer control (patent-pending) and packet delay and loss concealment HD audio handset with HD wideband audio codecs for excellent double-talk performance
Telephony Features	Intuitive graphic user interface (GUI), downloadable phone book (XML, LDAP), support for anonymous call using privacy header, MLS (multi language support) Voice mail indicator, downloadable custom ring-tones, call hold, call transfer (attended/blind), call forward, call waiting, caller ID, mute, redial, call log, caller ID display or block, Do-Not-Disturb (DND) and volume control 3-way conference, dial plan prefix, dial-plan support, off-hook auto dial, auto answer, early dial and speed dial

Network and Provisioning	Via keypad/LCD, Web browser, or secure (AES encrypted) central configuration file, manual or dynamic host configuration protocol (DHCP) network setup Support NAT traversal using IETF STUN and Symmetric RTP Support for IEEE 802.1p/Q tagging (VLAN), Layer 3 ToS
Firmware Upgrade	Support firmware upgrade via TFTP or HTTP Support for Authenticating configuration file before accepting changes User specific URL for configuration file and firmware files Mass provisioning using TR-069 or encrypted XML configuration file
Advanced Server Features	Message waiting indication, support DNS SRV Look up and SIP Server Fail Over, Support customizable idle screen via downloading XML by HTTP/TFTP
Security	User and administrator level passwords, MD5 and MD5-sess based authentication, AES based secure configuration file, SRTP, TLS, 802.1x media access control

Using the GXP1450 SIP Enterprise Phone

GETTING FAMILIAR WITH THE LCD

GXP1450 has a dynamic and customizable screen. The screen displays differently depending on whether the phone is idle or in use (active screen).

Table 7: LCD Display Definition

Item	Definitions
DATE AND TIME	Displays the current date and time. Can be synchronized with Internet time servers.
LOGO/NAME	Displays company logo/name. This logo/name can be customized via xml screen customization.
NETWORK STATUS	Displays the status of the phone and network. It will indicate whether the network is down, starting or running (IP address). "## MISSED CALLS" is shown here too.
STATUS ICON	Shows the status of the phone, using icons as shown in the next table.
LINE STATUS INDICATOR	Displays the name of the account that is in use. Select another account by pressing the LINE key on the left side.
SOFTKEYS	The softkeys are context sensitive and will change depending on the status of the phone. Typical functions assigned to soft-buttons are: • **FORWARD ALL** Unconditionally forwards the phone line to another phone. • **MISSED CALL** This option shows up there were unanswered calls to this phone. The Missed Calls option shows a list of the missed calls. • **NEXTSCR** Press this button to toggle between idle screen, weather and IP Address. • **REDIAL** Redials the last number. • **END CALL** Hangs up phone.

Table 8: LCD Icons

Icon	LCD Icon Definitions
	DND (idle): ON when "Do Not Disturb" is activated in idle screen

	DND (talking): ON when “Do Not Disturb” is activated in talking
	Forward All: INDICATES all incoming calls will be forwarded to the configured number
	Forward on Busy: INDICATES calls will be forwarded when phone is busy
	Forward on No Answer: INDICATES calls will be forwarded if the phone does not answer
	Forward All and No Answer: INDICATES calls will be forwarded if Forward All and Forward on No Answer are enabled
	Keypad Locked: ON when the keypad is locked
	Enter Keypad Password: Enter the keypad password to unlock the keypad
	Voice Mail: ON when there are new voice messages
	Network Status: Network is down
	Missed Call Icon: Indicates missed call(s)
	Save Call Record: Indicates phone system writing the call records into the flash. It might take 10 to 20 seconds to finish the process
	Waiting For Response: Please wait for the phone system to response before the keypad entry.
	Handset Mode
	Speaker Mode
	Headset Mode
	SRTP: SRTP is used during the call
	MUTE

FIGURE 1: GXP1450 KEYPAD LAYOUT

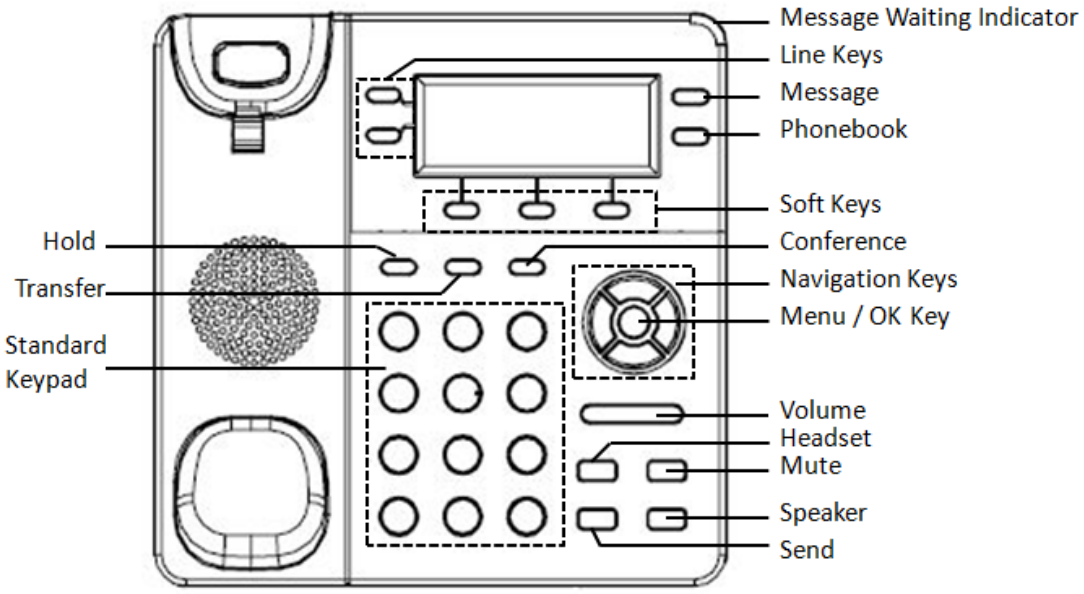


Table 9: GXP1450 Keypad Buttons

Key Button	Key Button Definitions
LINE KEYS	2 Line keys with LED, can be configured to different SIP profiles
HOLD	Place ACTIVE call on hold
TRANSFER	Transfer an ACTIVE call to another number
CONF	Press CONF button to connect Calling/Called party into conference
	Enter to retrieve voice mails or other messages
	Brings phonebook on screen
	Mute an active call
	Press HEADSET key to answer/hang up phone calls while using headset. It also allows user to toggle between headset and speaker
	Enable/Disable hands-free speaker
	Enable/Disable handset mode ; or used as SEND/REDIAL
	Press on "–" or "+" of the button to adjust the volume for handset/speakerphone/headset when the phone in idle or during call

	MENU/OK (The round button in the center): • Enter Keypad Configuration "MENU" mode when phone is in IDLE mode • Use as ENTER/OK key when in Keypad Configuration. Navigation keys "Up" "Down" "Left" and "Right" • Press to navigate in menu options • When the phone is idle, press "Up" to view missed call; press "Down" to view phonebook
0 - 9, *, #	Standard phone keypad; press # key to send call; press * key to for IVR functions

MAKING PHONE CALLS

Handset, Speakerphone and Headset Mode

The GXP1450 allows you to make phone calls via handset, speakerphone or headset mode. During the active calls the user can switch between the handset and the speaker by pressing the speaker key. For headset mode, the user must plug the headset to an RJ9 port on the back of the phone, which allows the user to pick up, speak or hang up calls.

Multiple SIP Accounts and Lines

GXP1450 can support up to two independent SIP accounts. Each account is capable of independent SIP server, user and NAT settings. Each of the line buttons is "virtually" mapped to an individual SIP account. The name of each account is conveniently printed next to its corresponding button. In off-hook state, select an idle line and the name of the account (as configured in the web interface) is displayed on the LCD and a dial tone is heard.

For example: Configure ACCOUNT 1 and ACCOUNT 2 with Account Name as "VoIP 1", "VoIP 2", respectively and ensure that they are active and registered. When LINE1 is pressed, you will hear a dial tone and see "VoIP 1" on the LCD display; when LINE2 is pressed, you will hear a dial tone and see "VoIP 2" on the LCD display.

To make a call, select the line you wish to use. The corresponding LINE LED will light up in green. User can switch lines before dialing any number by pressing the same LINE button one or more times. If you continue to press a LINE button, the selected account will circulate among the registered accounts.

For example: when LINE1 is pressed, the LCD displays "VoIP 1"; If LINE1 is pressed twice, the LCD displays "VoIP 2" and the subsequent call will be made through SIP account 2.

Incoming calls to a specific account will attempt to use its corresponding LINE if it is not in use. When the "virtually" mapped line is in use, the GXP1450 will flash the next available LINE in red. A line is ACTIVE when it is in use and the corresponding LED is red.

Completing Calls

There are five ways to complete a call:

1. **DIAL:** To make a phone call.
   - Take Handset/SPEAKER/Headset off-hook
     or press an available LINE key (activates speakerphone)
   - The line will have a dial tone and the primary line (LINE1) LED is red
     If you wish, select another LINE key (alternative SIP account)
   - Enter the phone number
   - Press the SEND key
     or press the "DIAL" softkey

2. **REDIAL:** To redial the last dialed phone number.
   When redialing the phone will use the same SIP account as was used for the last call. Thus, when the second SIP account was made for the last call/call attempt, the phone will use the second account to redial.
   - Take Handset/SPEAKER/Headset off-hook or
     press an available LINE key (activates speakerphone), the corresponding LED will be red
   - Press the SEND button
     or press the REDIAL softkey

3. **USING THE CALL HISTORY:** To call a phone number in the phone's history.
   When using the call history, the phone will use the same SIP account as was used for the last call/call attempt. Thus, when returning a call made to the second SIP account, the phone will use the second SIP account return the call.
   - Press the MENU button to bring up the Main Menu
   - Select Call History and then "Answered Calls", "Dialed Calls", "Missed Calls", "Transferred Calls" or "Forwarded Calls" depending on your needs
   - Select phone number using the arrow keys
   - Press OK to select
   - Press OK again to dial

4. **USING THE PHONEBOOK:** To call a phone in from the phone's phonebook.
   Each entry in the phonebook can be attached to an individual SIP account. The phone will use that SIP account to make the phone call.
   - Go to the phonebook by:
     i. Pressing the phonebook button (bottom, left-hand side of phone), or
     ii. Pressing the DOWN arrow key, or
     iii. Pressing the MENU/OK round button and
     Selecting "Phone Book" and
     Press MENU/OK again to enter the phonebook entries
   - Select the phone number by using the arrow keys
   - Press MENU/OK so select
   - Press the DOWN arrow key to select "Dial"
   - Press MENU/OK again to dial

5. **PAGING/INTERCOM:**
   The paging/intercom function can only be used if the SERVER/PBX supports this feature and both the phones and PBX are correctly configured.
   - Take the Handset/SPEAKER/Headset off-hook
   - Select the LINE key associated with account and dial tone will be heard
   - Press MENU/OK key to display LCD: **LINEx: PAGE**
   - Dial the phone number you want to Page/Intercom
   - Press SEND key

NOTE: Dial tone and dialed number display occurs after the phone is off-hook and the line key is selected. The phone waits 4 seconds (by default; No key Entry Timeout) before sending and initiating the call in offhook mode. Press the "SEND" (or "#" button if # is configured to be used as Dial) to override the 4 second delay.

Making Calls using IP Addresses

Direct IP calling allows two phones to talk to each other in an ad-hoc fashion without a SIP proxy. VoIP calls can be made between two phones if:

- Both phones have public IP addresses, or
- Both phones are on a same LAN/VPN using private or public IP addresses, or
- Both phones can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ)

To make a direct IP call, please follow these steps:

1. Press MENU button to bring up MAIN MENU
2. Select "Direct IP Call" using the arrow-keys
3. Press OK to select
4. Input the 12-digit target IP address. (Please see example below)
5. Press OK key to initiate call

To make a quick IP call, please refer to next section.

For example: If the target IP address is 192.168.1.60 and the port is 5062 (e.g. 192.168.1.60:5062), input the following: **192*168*1*60#5062 -** The " ***** " key represent the dot"**.**" ; The "#" key represent colon "**:**". Press OK to dial out.

Quick IP Call Mode

The GXP1450 also supports Quick IP call mode. This enables the phone to make direct IP-calls, using only the last few digits (last octet) of the target phone's IP-number.

This is possible only if both phones are in under the same LAN/VPN. This simulates a PBX function using the CMSA/CD without a SIP server. Controlled static IP usage is recommended.

Setting up the phone to make Quick IP calls

To enable Quick IP calls, the phone has to be setup first. This is done through the web-setup function. In the "Advanced Settings" page, set the "Use Quick IP-call mode to YES. When #*xxx* is dialed, where *x* is 0-9 and *xxx* <=255, a direct IP call to aaa.bbb.ccc.XXX is completed. "aaa.bbb.ccc" is from the local IP address *regardless* of subnet mask. The numbers #xx or #x are also valid. The leading 0 is not required (but OK).

For example:

192.168.0.2 calling 192.168.0.3 -- offhook the phone, dial #3 followed by SEND or #
192.168.0.2 calling 192.168.0.23 -- offhook the phone, dial #23 followed by SEND or #
192.168.0.2 calling 192.168.0.123 -- offhook the phone, dial #123 followed by SEND or #
192.168.0.2: dial #3 and #03 and #003 results in the same call -- call 192.168.0.3

NOTE: If you have a SIP Server configured, a Direct IP-IP still works. If you are using STUN, the Direct IP-IP call will also use STUN. Configure the "Use Random Port" to "NO" when completing Direct IP calls.

ANSWERING PHONE CALLS

Receiving Calls

1. **Incoming single call:** Phone rings with selected ring-tone. The corresponding account LINE flashes red. Answer call by taking Handset/SPEAKER/Headset off hook or pressing SPEAKER or by pressing the corresponding account LINE button.
2. **Incoming multiple calls:** When another call comes in while having an active call, the phone will produce a Call Waiting tone (stutter tone). Next available lines will flash in red (as described in section 4.3.2). Answer the incoming call by pressing its corresponding LINE button. The current active call will be put on hold.
3. **Paging/Intercom Enabled:** Phone beeps once and automatically establishes the call via SPEAKER. (PBX or Server must also supports this feature)

Do Not Disturb

1. Press the menu button, and scroll down to "Preference".
2. Select "Do Not Disturb" by pressing menu button.
3. Use arrow keys to either enable or disable "Do Not Disturb" feature.
4. When enabled, there will be a special "Do Not Disturb" icon appearing on the display. This will send the incoming caller directly to voicemail (if the callee has voicemail set up).

PHONE FUNCTIONS DURING A PHONE CALL

Call Waiting/ Call Hold

1. **Hold:** Place a call on hold by pressing the "HOLD" button.
2. **Resume:** Resume call by pressing the corresponding blinking LINE.
3. **Multiple Calls:** Automatically place ACTIVE call on hold by selecting another available LINE to place or receive another call. Call Waiting tone (stutter tone) audible when line is in use.

Mute

1. Press the MUTE button to enable/disable muting the microphone.
2. The "Line Status Indicator" will show "LINEx: SPEAKING" or "LINEx: MUTE" to indicate whether the microphone is muted.

Call Transfer

GXP1450 supports both *Blind* and *Attended* transfer. Also, users could make auto-attended transfer when this feature is enabled from web GUI.

1. **Blind Transfer:** Press "TRANSFER" button, then dial the number. Press the "SEND" button (or **#** key), or simply hang up the call to complete transfer of active call.

2. **Attended Transfer:** Press “LINEx” button to make a call and automatically place the ACTIVE LINE on HOLD. Once the call is established, press “TRANSFER” key then the LINE button of the waiting line to transfer the call. Hang up the phone call after the transfer is complete.
3. **Auto-Attended Transfer:** Users could enable Auto-Attended Transfer under Web GUI->Advanced Setting Page. During the first call, press “TRANSFER” hard button and it will bring up another line. The first call will be on hold. Enter the number and press SEND key to establish the second call (If pressing soft key “transfer” after entering the number, it will do blind transfer instead). After the second call is established, press “TRANSFER” hard button again. Now the phone will hang up and the call will be transferred.

NOTE:

- To transfer calls across SIP domains, SIP service providers must support transfer across SIP domains. Blind transfer will usually use the primary account SIP profile.

3-Way Conferencing

GXP1450 can host conference calls and supports up to 3-way conference calling.

1. **Initiate a Conference Call**:
   - Establish a connection with two parties
   - Press CONF button
   - Choose the desired line to join the conference by pressing the corresponding LINE button

2. **Cancel Conference:**
   - If after pressing the “CONF” button, a user decides not to conference anyone, press the current active LINE button
   - This will resume two-way conversation with the current line

3. **End Conference:**
   - Press HOLD to end the conference call and put all parties on hold
   - To speak with an individual party, select the corresponding blinking LINE

GXP1450 also supports **Easy Conference mode**. In Easy Conference mode, users can initiate conference by calling another number when the current line is in talking or conference. Also the conference can be re-established by pressing the ReConf softkey when the conference is on hold. Easy Conference mode can be used combined with the traditional ways to establish the 3-way conference above.

1. **Initiate a Conference Call:**
   - Establish one call
   - Press CONF button and a new line will be brought up
   - Dial the number and press SEND button to establish the second call
   - Press CONF button again or press the ConfCall softkey to establish the 3-way conference

2. **Join More Parties in Existed Conference:**
   - Establish conference call
   - Press CONF button and a new line will be brought up
   - Dial the number and press SEND button to establish the new call
   - Press CONF button again or press the ConfCall softkey to join the new party in the existed conference

3. **Hold Conference:**
   - During the conference, press HOLD button and the conference will be put on hold

- To resume the conference, press the ReConf softkey
- To split the conference and resume the call with each party, press the corresponding line key

4. **End Conference:**
   - If the users decide not to conference after establishing the second call, press EndCall softkey instead of ConfCall softkey/CONF button. It will end the second call and the screen will show the first call/conference is on hold.
   - During the conference, press EndCall softkey or hang up to end the conference

NOTE:

- The party that starts the conference call has to remain in the conference for its entire duration, you can put the party on mute but it must remain in the conversation. Also, this is not applicable when the feature "Transfer on call hangup" is turned on.
- When using Easy Conference mode, press SEND button to establish the second call after entering the number instead of using "#".

Voice Messages (Message Waiting Indicator)

A blinking red MWI (Message Waiting Indicator) indicates a message is waiting. Press the MSG button to retrieve the message. An IVR will prompt the user through the process of message retrieval. Press a specific LINE to retrieve messages for a specific line account.

NOTE:

- Each line has a separate voicemail account. Each account requires a voicemail portal number to be configured in the "voicemail user id" field.
- To check which line account has a message 1) press the message button (this always checks the primary account), 2) check each line for stutter tone or 3) check missed calls using the menu.

Shared Call Appearance (SCA)

The GXP1450 phone supports shared call appearance by Broadsoft standard. This feature allows members of the SCA group to shared SIP lines and provides status monitoring (idle, active, progressing, hold) of the shared line. When there is an incoming call designated for the SCA group, all of the members of the group will be notified of an incoming call and will be able to answer the call from the phone with the SCA extension registered.

All the users that belong to the same SCA group will be notified by visual indicator when a user seizes the line and places an outgoing call, and all the users of this group will not be able to seize the line until the line goes back to an idle state or when the call is placed on hold. (With the exception of when multiple call appearances are enabled on the server side)

In the middle of the conversation, there are two types of hold: Public Hold and Private Hold. When a member of the group places the call on public hold, the other users of the SCA group will be notified of this by the red-flashing button and they will be able to resume the call from their phone by pressing the line button. However, if this call is placed on private-hold, no other member of the SCA group will be able to resume that call.

To enable shared call appearance, the user would need to register the shared line account on one of the accounts on the phone. In addition, they would need to navigate to "Settings"->"Basic Settings" on the web GUI and set the line to "Shared Line" with the corresponding account. If the user requires more shared call appearances, the user can configure multiple line buttons to be "shared line" buttons associated with the account.

CALL FEATURES

The GXP1450 supports traditional and advanced telephony features including caller ID, caller ID w/name, call forward/transfer/park/hold as well as intercom/paging.

Table 10: GXP1450 Call Features

Key	Call Features
***30**	**Block Caller ID** (for all subsequent calls) Offhook and dial "*30".
***31**	**Send Caller ID** (for all subsequent calls) Offhook and dial "*31".
***67**	**Block Caller ID** (per call) Offhook, dial "*67" and then enter the number to dial out.
***82**	**Send Caller ID** (per call) Offhook, dial "*82" and then enter the number to dial out.
***70**	**Disable Call Waiting** (per Call) Offhook, dial "*70" and then enter the number to dial out.
***71**	**Enable Call Waiting** (per Call) Offhook, dial "*71" and then enter the number to dial out.
***72**	**Unconditional Call Forward** Offhook, dial "*72". Then enter the number to forward the call. Press OK softkey or SEND button.
***73**	**Cancel Unconditional Call Forward** Offhook, dial "*73" and the phone will hang up.
***90**	**Busy Call Forward** Offhook, dial "*90". Then enter the number to forward the call. Press OK softkey or SEND button.
***91**	**Cancel Busy Call Forward** Offhook, dial "*91" and the phone will hang up.
***92**	**Delayed Call Forward** Offhook, dial "*92". Then enter the number to forward the call. Press OK softkey or SEND button.
***93**	**Cancel Delayed Call Forward** Offhook, dial "*93" and the phone will hang up.

CUSTOMIZED LCD SCREEN & XML

GXP1450 Enterprise IP phone supports: 1) XML Custom Screen and 2) XML Downloadable Phonebook. Please refer to the following link for documentation and templates.

1) XML Custom Screen (custom idle screen logo, softkey layout and etc.)

http://www.grandstream.com/products/gxp_series/general/documents/GXP21xx_1450_XML_Screen_Customization.rar

2) XML Downloadable Phonebook

http://www.grandstream.com/products/gxp_series/general/documents/gxp_wp_xml_phonebook.pdf

Configuration Guide

The GXP1450 can be configured in two ways. Firstly, using the Key Pad Configuration Menu on the phone; secondly, through embedded web-configuration menu.

CONFIGURATION VIA KEYPAD

To enter the MENU, press the round button. Navigate the menu by using the arrow keys: up/down and left/right. Press the OK button to confirm a menu selection, delete an entry by pressing the MUTE/DEL button. The phone automatically exits MENU mode with an incoming call, the phone is off-hook or the MENU mode if left idle for 20 seconds.

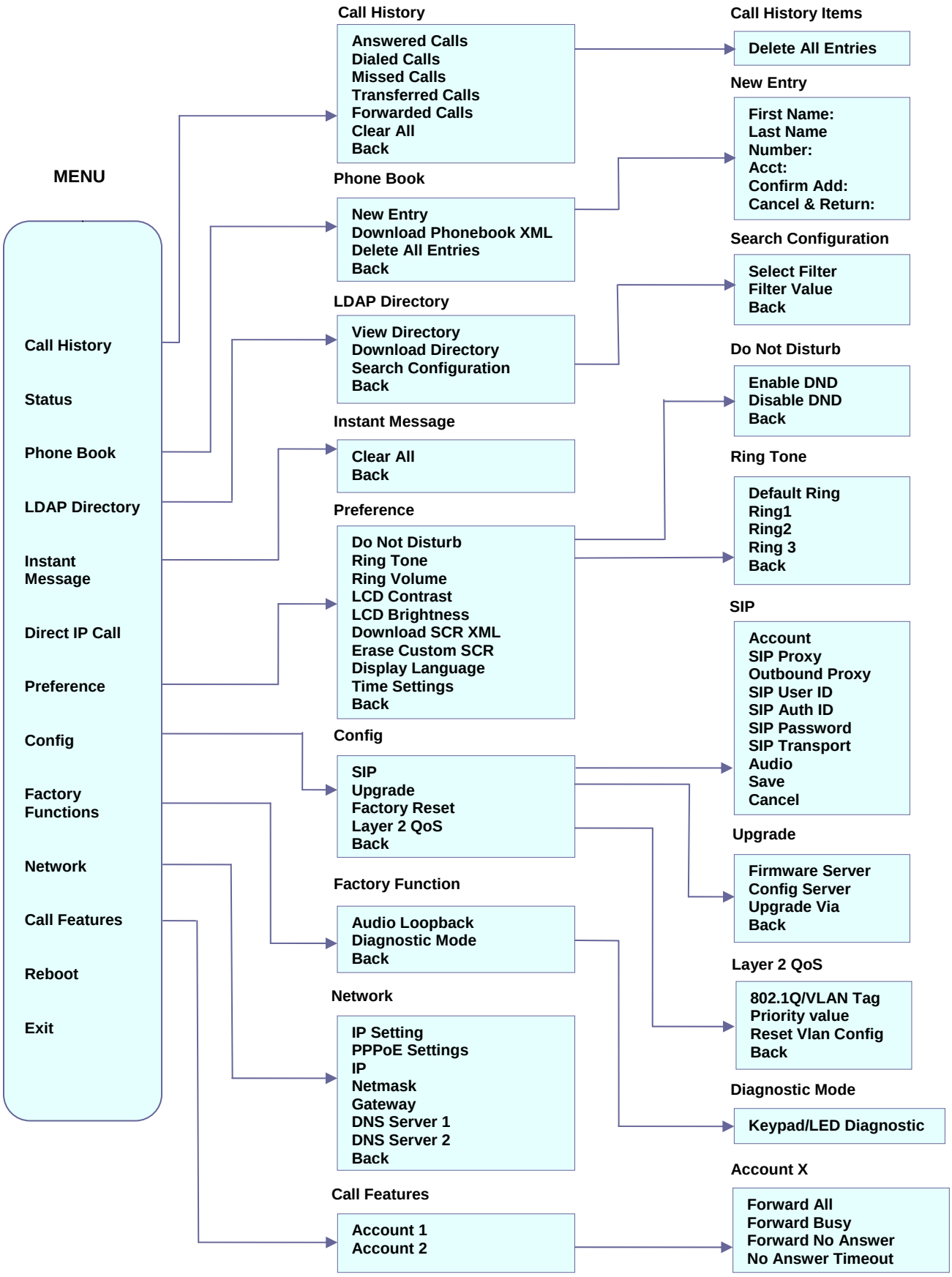
Press the MENU button to enter the Key Pad Menu. The menu options available are listed in table 11.

Table 11: Key Pad Configuration Menu

Item	Description
Call History	Displays histories of answered, dialed, missed, and transferred and forwarded calls
Status	Displays the network status, account status, software version, MAC-address and hardware version of the phone
Phone Book	Displays the phonebook and downloads phonebook XML
LDAP Directory	Displays the LDAP directory and downloads directory
Instant Messages	Goes to instant messages
Direct IP Call	Dials IP address for direct IP call
Preference	Press *Menu* button to enter this sub menu including: • **Do NOT Disturb** DND (Do Not Disturb) function could be turned on or off in the "Do Not Disturb" menu. • **Ring Tone** Choose different ring tones in the "Ring Tone" menu. • **Ring Volume** Press *Menu* button to hear the selected ring volume, press "←" or "→" to hear and adjust the ring tone volume. • **LCD Contrast** Press "←" or "→" to adjust the LCD contrast. • **LCD Brightness** Press "←" or "→" to adjust the LCD brightness for active/idle screen. • **Download SCR XML** The phone will download the custom idle screen if available. • **Erase Custom SCR** Custom idle screen will be erased and will be replaced with default logo. • **Display Language** Users can choose English, Simplified Chinese, Traditional Chinese, Korean, Japanese, Italian, Spanish, French, German, Portuguese, Russian, Croatian, Hungarian, Polish, Slovenian, Arabic, Hebrew or Dutch which are built in the phone. Users could select Automatic for

	local language based on IP location if available. Also, the phone will download secondary language if available. • **Time Settings** Users can set the date and time on the phone. Press *Menu* button to choose the menu item Press "←" or follow the soft keys to return to the main menu
Config	Press *Menu* button to display the configuration selections: • **SIP** To change SIP server settings for SIP account (SIP Proxy, Outbound Proxy, SIP User ID, SIP Auth ID, SIP Password, SIP Transport and Audio). • **Upgrade** To configure the firmware server and Config server for upgrading or provisioning the phone. • **Factory Reset** Key in the physical/MAC address on the back of the phone. Press OK softkey to reset to FACTORY DEFAULT setting. Do not use Factory Reset unless you want to restore factory settings. • **Layer 2 QoS** Configure 802.1Q/VLAN Tag and priority value.
Factory Functions	Press *Menu* to display the factory function items including • **Audio Loopback** Speak into the handset. If you hear your voice in the handset, your audio is working fine. Press *Menu* button to exit the mode. • **Diagnostic Mode** All LEDs will light up. Press any key on the keypad, to display the button name in the LCD. Lift and put back the handset or press *Menu* button to exit the diagnostic mode. Press '←' to return the main menu
Network	To select IP mode (DHCP/Static IP/PPPoE); to setup PPPoE, IP address, Netmask, Gateway address and DNS Server 1 and DNS Server 2.
Call Features	To enable/disable and configure Forward All, Forward Busy, Forward No Answer, No Answer Timeout, press Call Features and select the Account x to set the forward call features on this account.
Reboot	Select on Reboot and press *Menu* button to reboot the device.
Exit	Exit from this menu

FIGURE 2: KEYPAD GUI FLOW


MENU
Call History
Status
Phone Book
LDAP Directory
Instant Message
Direct IP Call
Preference
Config
Factory Functions
Network
Call Features
Reboot
Exit
Call History
Answered Calls
Dialed Calls
Missed Calls
Transferred Calls
Forwarded Calls
Clear All
Back
Phone Book
New Entry
Download Phonebook XML
Delete All Entries
Back
LDAP Directory
View Directory
Download Directory
Search Configuration
Back
Instant Message
Clear All
Back
Preference
Do Not Disturb
Ring Tone
Ring Volume
LCD Contrast
LCD Brightness
Download SCR XML
Erase Custom SCR
Display Language
Time Settings
Back
Config
SIP
Upgrade
Factory Reset
Layer 2 QoS
Back
Factory Function
Audio Loopback
Diagnostic Mode
Back
Network
IP Setting
PPPoE Settings
IP
Netmask
Gateway
DNS Server 1
DNS Server 2
Back
Call Features
Account 1
Account 2
Call History Items
Delete All Entries
New Entry
First Name:
Last Name
Number:
Acct:
Confirm Add:
Cancel & Return:
Search Configuration
Select Filter
Filter Value
Back
Do Not Disturb
Enable DND
Disable DND
Back
Ring Tone
Default Ring
Ring1
Ring2
Ring 3
Back
SIP
Account
SIP Proxy
Outbound Proxy
SIP User ID
SIP Auth ID
SIP Password
SIP Transport
Audio
Save
Cancel
Upgrade
Firmware Server
Config Server
Upgrade Via
Back
Layer 2 QoS
802.1Q/VLAN Tag
Priority value
Reset Vlan Config
Back
Diagnostic Mode
Keypad/LED Diagnostic
Account X
Forward All
Forward Busy
Forward No Answer
No Answer Timeout

CONFIGURATION VIA WEB BROWSER

The GXP1450 embedded Web server responds to HTTP/HTTPS GET/POST requests. Embedded HTML pages allow a user to configure the IP phone through a Web browser such as Microsoft's IE, Mozilla Firefox, Google Chrome.

Access the Web Configuration Menu

To access the phone's Web Configuration Menu

- Connect the computer to the same network as the phone[1]
- Make sure the phone is turned on and shows its IP address
- Start a Web browser on your computer
- Enter the phone's IP address in the address bar of the browser[2]
- Enter the administrator's password to access the Web Configuration Menu[3]

[1] The Web-enabled computer has to be connected to the same sub-network as the phone. This can easily be done by connecting the computer to the same hub or switch as the phone is connected to. In absence of a hub/switch (or free ports on the hub/switch), please connect the computer directly to the phone using the PC port on the phone.

[2] If the phone is properly connected to a working Internet connection, the phone will display its IP address. This address has the format: xxx.xxx.xxx.xxx, where xxx stands for a number from 0 to 255. You will need this number to access the Web Configuration Menu. For example, if the phone shows 192.168.0.60, please use "*http://192.168.0.60*" in the address bar of your browser.

[3] The default administrator password is "**admin**"; the default end-user password is "**123**".

NOTE:

- When changing any settings, always SUBMIT them by pressing "UPDATE" button on the bottom of the page. If, after having submitted some changes, more settings have to be changed, press the menu option needed.
- All the options under Basic Setting and Account Setting, and most of the options under Advanced Setting do not require reboot after submitting the changes. Under Advanced Setting, the parameters on network configuration require reboot after update.

Definitions

This section will describe the options in the Web configuration user interface. As mentioned, a user can log in as an administrator or end-user.

Functions available for the end-user are:

- **Status:** Displays the network status, account status, software version and MAC address of the phone, and service status.
- **Basic Settings:** Basic preferences such as date and time settings, multi-purpose keys and LCD settings can be set here.

Additional functions available to administrators are:

- **Advanced Settings:** To set advanced network settings, codec settings and XML configuration settings and etc.
- **Account:** To configure the SIP account.

Table 12: Device Configuration - Status

MAC Address	The device ID, in HEXADECIMAL format. This will be used for provisioning and is written on the label in the original box as well as on the label located on the back panel of the device.
IP Address	This field shows IP address of GXP1450.
Product Model	This field contains the product model information.
Part Number	This field contains the product part number.
Software Version	• **Prog**: This is the main firmware release number, which is always used for identifying the software (or firmware) system of the phone. • **Boot**: Booting code version number • **Core**: Core code version number • **Base**: Base code version number • **DSP**: DSP code version number • **Aux**: Aux code version number
System Up Time	This field shows system up time since the last reboot.
System Time	This field shows the current time on the phone system.
Registered	Indicates whether accounts are registered to the related SIP server.
PPPoE Link Up	Indicates whether the PPPoE connection is enabled (connected to a modem).
Service Status	• **GUI**: shows the GUI status: running or stopped • **Phone**: shows the phone status: running or stopped
Core Dump	Download core dump file for troubleshooting when necessary.

Table 13: Device Configuration – Settings/Basic Settings

End User Password	This contains the password for end user to access the Web Configuration Menu. Users can enter new password here. This field is case sensitive with a maximum length of 25 characters.
Confirm Password	Enter the end user password again as above to confirm new password.
IP Address	The GXP1450 operates in three modes: 1. **DHCP mode**: The GXP1450 acquires IP address from the first DHCP server it discovers on the LAN. The DHCP option is reserved for NAT router mode. In DHCP mode, all the field values for the Static IP mode are not used. 2. **PPPoE mode**: Set PPPoE account ID, PPPoE password and PPPoE service name for the GXP1450 to establish PPPoE sesstion. 3. **Static IP mode**: Configure IP address, Subnet Mask, Gateway, DNS Server 1, DNS Server 2 and Preferred DNS Server if Static IP mode is used.

802.1x Mode	This option allows the user to enable/disable 802.1x mode on the phone. The default value is disabled. To enable 802.1x mode, this field should be set to EAP-MD5. Once enabled, the user would be required to enter the following information below to be authenticated on the network: 1. Identity 2. MD5 Password
Line Keys x	This allows the user to configure each line key and enable Shared Call Appearance for the line. Options available for Key Mode are : 1. Line 2. Shared Line
Time Zone	This parameter controls the date/time display according to the specified time zone. If "Allow DHCP Option 2 to override Time Zone setting" is checked, the time zone will be overridden by the DHCP server.
Self-Defined Time Zone	This parameter allows the users to define their own time zone. The syntax is: **std offset dst [offset], start [/time], end [/time]** Default is set to: **MTZ+6MDT+5,M4.1.0,M11.1.0** **MTZ+6MDT+5,** This indicates a time zone with 6 hours offset with 1 hour ahead which is U.S central time. If it is positive (+) if the local time zone is west of the Prime Meridian (A.K.A: International or Greenwich Meridian) and negative (-) if it is east. **M4.1.0,M11.1.0** The 1st number indicates Month: 1,2,3.., 12 (for Jan, Feb, .., Dec) The 2nd number indicates the nth iteration of the weekday: (1st Sunday, 3rd Tuesday…) The 3rd number indicates weekday: 0,1,2,..,6(for Sun, Mon, Tues, … ,Sat) Therefore, this example is the DST which starts from the first Sunday of April to the 1st Sunday of November.
Weather Update	By default, "Enable Weather Update:" is set to "Yes". If set to "No", weather information will not display on the phone. Settings to customize the display of weather via: • City Code – Enter city code • Update Interval – Refresh time in minutes • Degree Unit – Select Automatic, Fahrenheit or Celsius Weather information is displayed on GXP1450 LCD when "Enable Weather Update" is set to "Yes" and pressing the "SwitchSCR" soft-key once.
LCD Backlight Brightness	Set the LCD brightness level for idle state and active state. Range from 0 to 8 where 0 means off and 8 means the brightest.
LCD Contrast	Set LCD contrast. Range from 0 to 20.

Date Display Format	LCD date display in the idle screen. The following formats are supported: • **yyyy-mm-dd**: 2011-11-17 • **mm-dd-yyyy**: 11-17-2011 • **dd-mm-yyyy**: 17-11-2011 • **dddd, MMMM dd**: Thursday, November 17 • **MMMM dd, dddd**: November 17, Thursday
Time Display Format	LCD time display in 12 hour or 24 hour format.
Disable in-call DTMF display	Default is “No”. This field is used to hide the keypad input during a call.
Disable Missed Call Backlight	Default is “No”. By default, LCD backlight will light up whenever there is a missed call.
Always Ring Speaker	Default is "No". This options is used when Headset is used on "Toggle Headset/Speaker" mode. If selected to "Yes", when the phone is in Headset mode, both headset and speaker will ring on incoming call.
HEADSET Key Mode	When headset is connected to GXP1450, users could use the HEADSET button in two different modes: • **Default Mode**: - When GXP1450 is in idle, press HEADSET button to offhook the phone and make calls by using headset. Headset icon will display on the top of the call screen in dialing/talking status - When there is an incoming call, press HEADSET button to pick the call by using headset - When there is an active call using headset, press HEADSET button to hang up the call - When Speaker/Handset is used in dialing/talking status, press HEADSET button to switch to headset. Press it again to hang up the call, or press Speaker/Handset to switch back to previous mode • **Toggle Headset/Speaker** - When GXP1450 is in idle, press HEADSET button to switch to Headset mode. The idle screen will display a Headset icon. In this mode, if pressing Speaker button or softkey, headset will be used by default -When there is an incoming call, press LINE button, “Answer” softkey or Speaker button, headset will be used - -When there is an active call, press HEADSET button to toggle between Headset and Speaker
Headset TX gain (dB)	Set headset TX gain to -6, 0 or +6. Default is 0 db.
Headset RX gain (dB)	Set headset RX gain to -6, 0 or +6. Default is 0 db.
Handset TX gain (dB)	Set handset TX gain to -6, 0 or +6. Default is 0 db.

Table 14: Device Configuration – Settings /Advanced Settings

Admin Password	Administrator password. Only the administrator can access the "Advanced Settings" and "Account Settings" page. Password field is purposely blank for security reasons after clicking update and saved. The maximum password length is 25 characters.
Confirm Password	Enter the end user password again as above to confirm new password.
Layer 3 QoS	This field defines the layer 3 QoS parameter. It is the value used for IP Precedence or Diff-Serv or MPLS. Default value is 12.
Layer 2 QoS	This contains the value used for layer 2 802.1Q/VLAN tag and 802.1p priority value. Default setting is 0. VLAN supported equipment is required when configuring these settings.
Local RTP port	This parameter defines the local RTP port pair used to listen and transmit. It is the base RTP port for channel 0. When configured, channel 0 will use this port _value for RTP; channel 1 will use port_value+2 for RTP. Local RTP port ranges from 1024 to 65400 and must be even. The default value is 5004.
Use Random Port	This parameter, when set to "Yes", will force random generation of both the local SIP and RTP ports. This is usually necessary when multiple GXPs are behind the same NAT. Default is "No".
Keep-alive interval	This parameter specifies how often the GXP1450 sends a blank UDP packet to the SIP server in order to keep the "hole" on the NAT open. Default is 20 seconds.
Use NAT IP	NAT IP address used in SIP/SDP message. Default is blank.
STUN Server	IP address or Domain name of the STUN server. STUN resolution result will display in the STATUS page of the Web UI.
Firmware Upgrade and Provisioning	Allows the user to select the following options for firmware upgrade: • Always Check for New Firmware • Check New Firmware only when F/W pre/suffix changes • Always Skip the Firmware Check. Firmware upgrade may take up to 10 minutes depending on network environment. Do not interrupt the firmware upgrading process. **Note:** Grandstream strongly recommends that the user upgrade firmware locally in a LAN environment if using TFTP to upgrade. Please DO NOT interrupt the upgrade process (especially the power supply) as this will damage the device.
XML Config File Password	The password used for encrypting the XML configuration file using OpenSSL. This is required for the phone to decrypt the encrypted XML configuration file.
HTTP/HTTPS User Name	The user name for the HTTP/HTTPS server.
HTTP/HTTPS Password	The password for the HTTP/HTTPS server.

Upgrade Via	This field allows the user to choose the firmware upgrade/config server path method: TFTP, HTTP or HTTPS. • **TFTP**: GXP1450 retrieves the new firmware files or new configuration file from the specified TFTP server path at boot time. If there is no new firmware file or configuration file, the system will start the boot process using the existing firmware or config file. If a TFTP server is configured and new firmware files are retrieved, the new downloaded image is saved into the Flash memory. (Please do NOT interrupt the TFTP upgrade process (especially the power supply) as this will damage the device. • **HTTP**: GXP1450 retrieves the new firmware files or new configuration file from the specified URL or IP for the HTTP server. For example: provisioning.mycompany.com:6688/Grandstream/1.0.0.6 **Note:** ":6688" is the specific TCP port where the HTTP server is listening; Omit if using default port 80. • **HTTPS**: GXP1450 retrieves the new firmware files or new configuration file from the specified URL or IP for the HTTP server via a secured HTTP connection. For example: provisioning.mycompany.com **Note:** HTTPS default port is 443.
Firmware Server Path	Defines the server path for the firmware server. It can be different from the Configuration server which is used for provisioning. For example: firmware.mycompany.com:6688/Grandstream/1.0.0.6
Config Server Path	Defines the server path for provisioning; it can be different from the firmware server. For example: provisioning.mycompany.com:6688/Grandstream/gxp1450
Firmware File Prefix/Postfix	Default is blank. If configured, GXP1450 will request the firmware file with the prefix/postfix and only the firmware with the matching encrypted prefix will be downloaded and flashed into the phone. **Note:** This setting is useful for ITSPs. End user should keep it blank.
Config File Prefix/Postfix	Default is blank. If configured, GXP1450 will request the config file with the prefix/postfix and only the file with the matching encrypted prefix will be downloaded and flashed into the phone. **Note:** This setting is useful for ITSPs. End user should keep it blank.

Allow DHCP Option 43 and Option 66 to override server	Default is “Yes”. This allows device gets provisioned from the server automatically.
Automatic Upgrade	Default is “No”. Choose “Yes” to enable automatic HTTP upgrade and provisioning. In “Check for upgrade every” field, enter the number of minutes to check the HTTP server for firmware upgrade or configuration changes. When set to “No”, the phone will only perform HTTP upgrade and configuration check once at boot up. **Note:** This function is used by ITSP. End user should NOT touch these parameters.
Authenticate Conf File	Default is “No”. If set to “Yes”, configuration file would be authenticated before acceptance. End user should use default setting.
Enable TR-069	Default is “No”.
ACS URL	URL for TR-069 Auto Configuration Servers (ACS).
TR-069 Username	Enter ACS username for TR-069.
TR-069 Password	Enter ACS password for TR-069.
Periodic Inform Enable	Enable periodic inform. Default is “No”. If set to YES, device will send inform packets to the ACS.
Periodic Inform Interval	When enabling periodic inform, set up the periodic inform interval to send the inform packets to the ACS.
Connection Request Username	Enter the connection request username. This is the user name for the ACS to connect to this device.
Connection Request Password	Enter the connection request password. This is the password for the ACS to connect to this device.
Authentication Method	Select the authentication method among “No authentication”, “Basic” or Digest.
Connection Request Port	Enter the connection request port.
Phonebook XML Download	Selects the file download mode for the download server. Users can choose from TFTP/HTTP/No.
Phonebook XML Server Path	The URL/IP address of the phonebook download server.
Phonebook Download Interval	The interval at which the phonebook will be downloaded from the download server (in Minutes). The default setting is 0.
Remove Manually-edited entries on Downloads	If set to “Yes”, the phone will remove the manually-edited entries in the old phonebook list before downloading the new file. The default setting is set to “Yes”. **Note:** If there is a duplicate entry (same name, same number and same index) added both manually and from phonebook xml file, only 1 entry will be saved and displayed.

LDAP Directory	IP address or domain name of LDAP script server. **Note:** When LDAP Directory is used, please set "Phonebook XML Download" to Yes to enable it.
Idle Screen XML Download	Enable XML Idle Screen download via TFTP or HTTP. Select whether to "Use Custom Filename" or not, and define the "XML server path".
Download Screen XML At Boot-up	The phone will download the idle screen xml file if set to "Yes". The default setting is "No".
Use custom filename	The phone will use custom filename specified in XML server path if set to "Yes". The default setting is "No".
Idle Screen XML Server Path	Specify the idle screen XML server path.
Offhook Auto Dial	To configure a User ID/extension to dial automatically when the phone is taken offhook.
Syslog Server	The IP address or URL of System log server. This feature is especially useful for ITSPs.
Syslog Level	Select the syslog level for GXP1450 to report. Default is NONE. The level is one of DEBUG, INFO, WARNING or ERROR. Syslog messages are sent based on the following events: • product model/version on boot up (INFO level) • NAT related info (INFO level) • sent or received SIP message (DEBUG level) • SIP message summary (INFO level) • inbound and outbound calls (INFO level) • registration status change (INFO level) • negotiated codec (INFO level) • Ethernet link up (INFO level) • SLIC chip exception (WARNING and ERROR levels) • memory exception (ERROR level) The Syslog uses USER facility. In addition to standard Syslog payload, it contains the following components: *GXP_1450: [device MAC address][firmware version]* error message. *For example*: Nov 30 07:55:46 10.131.28.27 GXP1450_GUI:[00:0b:82:2b:3c:a4][1.0.1.108][-LCD-] IdleScreen/XmlApp is now visible
Send SIP Log	When setting the "Yes", phone will send out SIP Log to syslog server. Default setting is "No".
NTP server	This parameter defines the URI or IP address of the NTP (Network Time Protocol) serve. It is used to display the current date/time.
Allow DHCP Option 42 to override NTP server	Default is "Yes". This allows device gets provisioned for DHCP Option 42 from the server automatically.

SSL Certificate	The user specified SSL certificate used for SIP over TLS.
SSL Private Key	The user specified SSL private key used for SIP over TLS.
SSL Private Key Password	User specified password to protect the private key above.
Distinctive Ring Tone	Select a Distinctive Ring Tone 1 through 3 for the caller ID. The Caller ID can be configured as a particular caller ID or Alert-Info text. When a particular call ID is configured in the Caller ID, the selected ring tone will be used when the incoming call is from the Caller ID. System Ring Tone will be used for all other calls. If server supports Alert-Info, the Alert-Info text can be mapped to the 3 customized ring tones. For example, if you configure the custom ring tone 1 user ID to “priority”, that ring tone will be used if we receive INVITE with Alert-Info header in the following format: Alert-Info:;info=priority
System Ring Tone	System ring tone. Default is North American standard. Users could adjust system ring tone frequencies and cadences based on local telecom standard.
Call Progress Tones	Using these settings, users can configure ring or tone frequencies based on parameters from local telecom. By default, they are set to North American standard. Frequencies should be configured with known values to avoid uncomfortable high pitch sounds. **Syntax**: f1=val,f2=val[,c=on1/off1[-on2/off2[-on3/off3]]]; - Frequencies are in Hz - Cadence on and off are in 10ms - ON is the period of ringing (“On time” in “ms”) while OFF is the period of silence - In order to set a continuous ring, OFF should be zero. Otherwise it will ring ON ms and a pause of OFF ms and then repeat the pattern - Up to three cadences are supported
Disable Call Waiting	Default is “No”. If set to “Yes”, the call waiting feature will be disabled.
Disable Call Waiting Tone	Default is “No”. If set to “Yes”, the call waiting tone will be disabled.
Disable Direct IP Calls	Default is “No”. If set to “Yes”, direct IP calls will be disabled.
Use Quick IP Call Mode	Dial an IP address under the same LAN/VPN segment by entering the last octet in the IP address. In the Advanced Settings page there is an option “Use Quick IP-call mode”. Default setting is “No”. When set to “Yes”, offhook the phone and dial #XXX (X is 0-9 and XXX <=255), phone will make direct IP call to aaa.bbb.ccc.XXX where aaa.bbb.ccc comes from the local IP address REGARDLESS of subnet mask. #XX or #X are also valid so leading 0 is not required (but OK). See **Quick IP Call Mode** for details.

Disable Conference	Default is “No”. If set to “Yes”, conference will be disabled.
Disable DND Button	Default is “No”. If set to “Yes”, the “DND” button on keypad will be disabled.
Disable Transfer	Default is “No”. If set to “Yes”, transfer will be disabled.
Auto-Attended Transfer	Default is “No”. If set to “Yes”, the phone will use attended transfer by default.
Configuration via Keypad Menu	Configures the access control of configurations via the phone keypad menu. There are three modes: • Unrestricted • Basic Settings Only: CONFIG option will not display in keypad MENU • Constraint Mode: • CONFIG, FACTORY FUNCTIONS and NETWORK options will not display in keypad MENU
Enable STAR key Keypad locking	Default is enabled. When the phone is in idle screen, press and hold STAR key for 4 seconds and the keypad will be locked. To unlock the keypad, press and hold STAR key for 4 seconds. Enter the password to unlock in the prompted window, and then press MENU key to enter. The password to lock/unlock can be configured. If users forget the password to unlock it, log into web GUI and change the password. Update and reboot the phone. When the phone boots up, press and hold STAR key for 4 seconds and enter the new password to unlock it.
Password to lock/unlock	Enter the password to lock the keypad in web GUI. To unlock the keypad, enter the password in the prompted window in the phone’s LCD screen.
Do not escape “#” as %23 in SIP URI	Default is “No”. By default, # will be replaced as %23 in SIP URI.

Display Language	Allows user to choose preferred display language in web UI and keypad UI. Currently, the phone supports these languages: Arabic, German, English, Spanish, French, Hebrew, Croatian, Hungarian, Italian, Japanese, Korean, Dutch, Polish, Portuguese, Russian, Slovenian, Simplified Chinese and Traditional Chinese. **Note:** The "Automatic" setting in language refers to Grandstream's IP2Location client which when connected to Internet would attempt to lookup a database (driven by Grandstream) with the IP address for its geographical location. Language file postfix allows the language file to have different postfixes so the phone can request a particular file. It will append an underscore "_" plus the string in the language file postfix. The default language file name is "gxp.txt". If the field "Language File postfix "has "NL" string in it, then the phone will request "gxp_NL.txt" instead of "gxp.txt". User can only load one secondary language. Supported downloadable language: Czech, Croatian, Estonian, French, German, Italian, Polish, Portuguese, Slovak, Slovenian and Spanish. How to set up Download Language: This is similar to updating firmware in your local network environment. 1. Get the language file gxp.txt ready. Make sure the file is using UTF-8 encoding. 2. Copy gxp.txt to the firmware server directory using your local TFTP or HTTP server. 3. Access the advanced settings of the Web GUI, set "Display Language" to "Download Language" and enter the server path in Firmware Server Path. Select TFTP or HTTP for firmware upgrade. 4. Update and reboot the phone.
Download Device Configuration	Download the current device configuration txt file. In the txt file, all the P values will be displayed except for the password fields.

GXP1450 has up to two line appearances, each with an independent SIP account. Each SIP account requires its own configuration page. Their configurations are identical.

Table 15: SIP Account Settings

Account Active	This field indicates whether the account is active. The default value is "Yes".
Account Name	The name associated with each account - displayed on LCD.
SIP Server	SIP Server's IP address or Domain name provided by VoIP service provider.
Secondary SIP Server	This field allows administrator to configure a backup SIP Server.
Outbound Proxy	IP address or Domain name of Outbound Proxy, Media Gateway, or Session Border Controller. Used for firewall or NAT penetration in different network environment. If the system detects symmetric NAT, STUN will not work. ONLY outbound proxy can provide solution for symmetric NAT.

SIP User ID	User account information provided by VoIP service provider (ITSP); either an actual phone number or formatted like one.
Authenticate ID	SIP service subscriber's Authenticate ID used for authentication. It can be identical to or different from SIP User ID.
Authenticate Password	SIP service subscriber's account password for GXP1450 to register to (SIP) servers of ITSP.
Name	SIP service subscriber's name that is used for Caller ID display.
DNS Mode	The default is set to A Record. If user wishes to locate the server by DNS SRV, the user may select SRV or NATPTR/SRV. When "Use Configured IP" option is selected, if SIP server is configured as domain name, phone will not send DNS query, but use "Primary IP" or "Secondary IP" to send sip message if at least one of them are not empty.
Primary IP	This option applies only if "Use Configured IP" is selected, the phone will send DNS query to the Primary IP. Insert IP address here.
Backup IP 1	Insert the first back up IP here.
Backup IP 2	Insert the second back up IP here.
TEL URI	Default is "Disabled". If the phone has an assigned PSTN telephone number, this field should be turned on and a "User=Phone" parameter will be attached to the "From" header in SIP request.
SIP Registration	This parameter controls sending REGISTER messages to the proxy server. The default setting is "Yes".
Unregister on Reboot	Default is "No". If set to "Yes", the SIP user's registration information will be cleared on reboot.
Register Expiration	This parameter allows user to specify the time frequency (in minutes) that GXP1450 refreshes its registration with the specified registrar. The default interval is 60 minutes. The maximum interval is 65,535 minutes (about 45 days).
Reregister Before Expiration	This parameter allows user to specify the time frequency (in seconds) that GXP1450 sends out a re-registration request before the Register Expiration. By default is 0 second.
Local SIP Port	This parameter defines the local SIP port used to listen and transmit. The default value for Account 1 is 5060 and 5062 for Account 2.
SIP Registration Failure Retry Wait Time	Retry registration if the process failed. Default is 20 seconds.
SIP T1 Timeout	RFC 3261 SIP T1 timer. Default is 0.5 second.
SIP T2 Interval	RFC 3261 SIP T2 timer. Default is 4 seconds.
SIP Transport	Choose SIP Transport between UDP and TCP. Default is UDP.
SIP URI Scheme when using TLS	Select "sip:" or "sips:". Default is "sips:".

Use Actual Ephemeral Port in Contact with TCP/TLS	Enable to use actual ephemeral port in contact with TCP/TLS. Default is “No”.
Check Domain Certificate	Enable to check the domain certificate. Default is “No”.
Remove OBP from Route	The SIP Extension notifies the SIP server that it is behind a NAT/firewall.
Validate Incoming Messages	This configuration selects whether or not the incoming messages should be validated.
Support SIP Instance ID	Selects whether or not SIP Instance ID is supported.
NAT Traversal	This parameter activates the NAT traversal mechanism. It has options: No, STUN, Keep-Alive, UPnP, Auto, VPN. If selecting STUN and a STUN server is also specified, the phone performs according to the STUN client specification. Using this mode, the embedded STUN client detects if and what type of NAT/Firewall configuration is used. If the detected NAT is a Full Cone, Restricted Cone, or a Port-Restricted Cone, the phone will use its mapped public IP address and port in all of its SIP and SDP messages. If selecting Keep-Alive with no specified STUN server, the GXP1450 will periodically (every 20 seconds or so) send a blank UDP packet (with no payload data) to the SIP server to keep the “hole” on the NAT open.
SUBSCRIBE for MWI	Default is “No”. When set to “Yes” a SUBSCRIBE for Message Waiting Indication will be sent periodically.
SUBSCRIBE for Registration	Default is “No”. When set to “Yes” a SUBSCRIBE for Registration will be sent periodically.
Feature Key Synchronization	Default is “No”. This option is to synchronize DND/Call Forward features with Broadsoft platform. When set to “Yes”, a SUBSCRIBE will be sent out periodically to the server. Then when DND/Call Forward features (Call Forward No Answer, Unconditional Call Forward and Call Forward on Busy) are configured or changed on the phone and Broadsoft server side, those features will be synchronized on the phone side and Broadsoft server side.
Proxy-Require	SIP Extension to notify SIP server that the unit is behind the NAT/Firewall.
Voice Mail UserID	When configured, user can access messages by pressing “MSG” button. This ID is usually the VM portal access number.

Send DTMF	This parameter specifies the mechanism to transmit DTMF digit. There are 3 supported modes: - In audio: DTMF is combined in audio signal (not very reliable with low-bit-rate codec) - via RTP (RFC2833) - via SIP INFO.
DTMF Payload Type	Sends DTMF using RFC2833. The default is 101.
Early Dial	Default is “No”. Use only if proxy supports 484 responses.
Dial Plan Prefix	Sets the prefix added to each dialed number.
Dial Plan	Dial Plan Rules: 1. Accepted Digits: 1,2,3,4,5,6,7,8,9,0 , *, #, A,a,B,b,C,c,D,d 2. Grammar: x - any digit from 0-9; a) xx+ - at least 2 digit numbers b) xx. - only 2 digit numbers c) ^ - exclude d) [3-5] - any digit of 3, 4, or 5 e) [147] - any digit of 1, 4, or 7 f) <2=011> - replace digit 2 with 011 when dialing g) | - the OR operand • Example 1: {[369]11 | 1617xxxxxxx} Allow 311, 611, and 911 or any 10 digit numbers with leading digits 1617 • Example 2: {^1900x+ | <=1617>xxxxxxx} Block any number of leading digits 1900 or add prefix 1617 for any dialed 7 digit numbers • Example 3: {1xxx[2-9]xxxxxx | <2=011>x+} Allows any number with leading digit 1 followed by a 3 digit number, followed by any number between 2 and 9, followed by any 7 digit number OR Allows any length of numbers with leading digit 2, replacing the 2 with 011 when dialed. 3. Default: Outgoing – {x+} Allow any length of numbers. Example of a simple dial plan used in a Home/Office in the US: { ^1900x. | <=1617>[2-9]xxxxxx | 1[2-9]xx[2-9]xxxxxx | 011[2-9]x. | [3469]11 } Explanation of example rule (reading from left to right): • ^1900x. - prevents dialing any number started with 1900 • <=1617>[2-9]xxxxxx - allows dialing to local area code (617) numbers by dialing 7 numbers and 1617 area code will be added automatically • 1[2-9]xx[2-9]xxxxxx |- allows dialing to any US/Canada Number with 11 digits length • 011[2-9]x. - allows international calls starting with 011 • [3469]11 - allow dialing special and emergency numbers 311, 411, 611 and 911 **Note:** In some cases where the user wishes to dial strings such as *123 to activate voice mail or other applications provided by their service provider, the * should be predefined inside the dial plan feature. An example dial plan will be: { *x+ } which allows the user to dial * followed by any length of numbers.
Delayed Call Forward Wait Time	Time waited before the call is forward to a number or VM. Default is 20 seconds.

Enable Call Features	Default is “Yes”. If set to “No”, Call transfer, Call Forwarding & Do-Not-Disturb are supported locally provided ITSP support those features. In addition, “ForwardAll” softkey will be hidden if call feature code is disabled for Account 1.
Call Log	Enable/disable Call Log and select type of calls to log: - Log All Calls - Log Incoming/Outgoing only (Missed calls NOT recorded) - Disable Call Log
Session Expiration	The SIP Session Timer extension enables SIP sessions to be periodically “refreshed” via a SIP request (UPDATE, or re-INVITE. Once the session interval expires, if there is no refresh via a UPDATE or re-INVITE message, the session is terminated. Session Expiration is the time (in seconds) at which the session is considered timed out, provided no successful session refresh transaction occurs beforehand. The default value is 180 seconds.
Min-SE	Defines the minimum session expiration (in seconds). Default is **90 seconds**.
Caller Request Timer	If set to “Yes”, the phone will use session timer when it makes outbound calls if remote party supports session timer.
Callee Request Timer	If selecting “Yes”, the phone will use session timer when it receives inbound calls with session timer request.
Force Timer	If set to “Yes”, the phone will use session timer even if the remote party does not support this feature. If set to “No”, the session timer is enabled only when the remote party supports this feature. To turn off Session Timer, select “No” for Caller Request Timer, Callee Request Timer, and Force Timer.
UAC Specify Refresher	As a Caller, select UAC to use the phone as the refresher, or UAS to use the Callee or proxy server as the refresher.
UAS Specify Refresher	As a Callee, select UAC to use caller or proxy server as the refresher, or UAS to use the phone as the refresher.
Force INVITE	Session Timer can be refreshed using INVITE method or UPDATE method. Select “Yes” to use INVITE method to refresh the session timer.
Enable 100rel	PRACK (Provisional Acknowledgment) method enables reliability to SIP provisional responses (1xx series). This is required to support PSTN inter-networking.
Account Ring Tone	There are 4 uniquely defined ring tones: • System Ring Tone: when selected, all calls will ring with system ring tone. • 3 Customer Ring Tones: when selected, incoming calls from designated account will play selected ring tone.
Ring Timeout	Defines how long ring will ring when receiving a call. Default is 60 seconds.
Line-seize Timeout	Defines how long it will take before seizing the line in Shared Call Appearance group. Default is 15 seconds.

Send Anonymous	If this parameter is set to “Yes”, the “From” header in outgoing INVITE message will be set to anonymous, essentially blocking the Caller ID from displaying.
Anonymous Call Rejection	Default is “No”. If set to “Yes”, anonymous call will be rejected.
Auto Answer	Default is “No”. If set to “Yes”, GXP1450 will automatically switch on speaker to answer the incoming call.
Allow Auto Answer by Call-Info	If the Call-Info header contains answer-after=0, the call be answered automatically. This fields need to be set to Yes if users would like to have the phone to be paged/intercom.
Refer-To Use Target Contact	Default is “No”. If set to “Yes”, then for Attended Transfer, the “Refer-To” header uses the transferred target’s Contact header information.
Transfer on Conference Hangup	Defines whether or not the call is transferred to the other party if the initiator of the conference hangs up. Default setting is set to “No”.
Check SIP User ID for Incoming INVITE	Check the SIP User ID in Request URI. If they don’t match, the call will be rejected.
Preferred Vocoder	GXP1450 supports up to 7 different Vocoder types including G.711(a/μ) (also known as PCMU/PCMA), G.723.1, G.729A/B, G.726-32, iLBC, G.722 (wide-band). Configure Vocoders in a preference list that is included with the same preference order in SDP message. Enter the first Vocoder in this list by choosing the appropriate option in “Choice 1”. Similarly, enter the last Vocoder in this list by choosing the appropriate option in “Choice 8”.
SRTP Mode	Enable SRTP mode based on selection. Default is “No”.
Symmetric RTP	Selects whether or not symmetric RTP is supported.
Silence Suppression	This controls the silence suppression/VAD feature of the audio codec G.723 and G.729. If set to “Yes”, when silence is detected, a small quantity of VAD packets (instead of audio packets) will be sent during the period of no talking. If set to “No”, this feature is disabled.

Voice Frames per TX	This field contains the number of voice frames to be transmitted in a single Ethernet packet (be advised the IS limit is based on the maximum size of Ethernet packet is 1500 byte (or 120kbps)). When setting this value, be aware of the requested packet time (ptime, used in SDP message) is a result of configuring this parameter. This parameter is associated with the first codec in the above codec Preference List or the actual used payload type negotiated between the 2 conversation parties at run time. *E.g.*, if the first codec is configured as G.723 and the "Voice Frames per TX" is set to 2, then the "ptime" value in the SDP message of an INVITE request will be 60ms because each G.723 voice frame contains 30ms of audio. Similarly, if this field is set to 2 and the first codec is G.729 or G.711 or G.726, then the "ptime" value in the SDP message of an INVITE request will be 20ms. If the configured voice frames per TX exceeds the maximum allowed value, the IP phone will use and save the maximum allowed value for the corresponding first codec choice. The maximum value for PCM is 10 (x10ms) frames; for G.726, it is 20 (x10ms) frames; for G.723, it is 32 (x30ms) frames; for G.729/G.728, 64 (x10ms) and 64 (x2.5ms) frames respectively. Please be careful when editing these parameters. Adjusting these parameters will also change the dynamic jitter buffer. The GXP1450 has a patent dynamic jitter buffer handling algorithm. The jitter buffer range is 20 ~ 200 ms. We recommend using the default settings provided. We do not recommend adjusting these parameters if you are an average user. Incorrect settings will affect the voice quality.
No Key Entry Timeout	Default is 4 seconds. After the timeout, the phone will send out the dialed number.
Use # as Dial Key	This parameter allows users to configure the "#" key as the "Send" (or "Dial") key. If set to "Yes", the "#" key will immediately send the call. In this case, this key is essentially equivalent to the "(Re)Dial" key. If set to "No", the "#" key is included as part of the dial string.
G723 Rate	Encoding rate for G723 codec. By default, 6.3kbps rate is set.
G726-32 Packing Mode	Select "ITU" or "IETF" for G726-32 packing mode.
iLBC Frame Size	iLBC packet frame size. Default is 20ms. For Asterisk PBX, 30ms might be required.
iLBC Payload Type	Payload type for iLBC. Default value is 97. The valid range is between 96 and 127.
Jitter Buffer Type	Jitter buffer type: Fixed or Adaptive. Default value is Adaptive.
Jitter Buffer Length	Jitter buffer length. Default value is 300ms. The valid range is between 100ms and 800ms.
Conference URI	Configure the conference URI when using Broadsoft N-way calling feature.
Special Feature	Default is Standard. Choose the selection to meet special requirements from Soft Switch vendors.

SAVING THE CONFIGURATION CHANGES

After the user makes a change to the configuration, press the “Update” button in the Configuration Menu. The web browser will then display a message window to confirm saved changes.

We recommend rebooting or powering cycle the IP phone after saving changes.

REBOOTING THE PHONE REMOTELY

Press the “Reboot” button at the bottom of the configuration menu to reboot the phone remotely. The web browser will then display a message window to confirm that reboot is underway. Wait 30 seconds to log in again.

Software Upgrade & Customization

Software (or firmware) upgrades are completed via either TFTP or HTTP. The corresponding configuration settings are in the ADVANCED SETTINGS configuration page.

FIRMWARE UPGRADE THROUGH TFTP/HTTP

To upgrade via TFTP or HTTP, select TFTP or HTTP upgrade method. "Upgrade Server" needs to be set to a valid URL of a HTTP server. Server name can be in either FQDN or IP address format. Here are examples of some valid URLs.

- firmware.mycompany.com:6688/Grandstream/1.2.3.5
- 72.172.83.110

There are two ways to set up the Upgrade Server to upgrade firmware: via Key Pad Menu or Web Configuration Interface.

Key Pad Menu

To configure the Upgrade Server via Key Pad Menu options, select "Config" from the Main Menu, then select "Upgrade". Under this sub Menu, user can edit Upgrade Server in either an IP address format or FQDN format. Choose "*Save and use TFTP*" or "*Save and use HTTP*" to select upgrade method. Select "Reboot" from the Main Menu to reboot the phone.

Web Configuration Interface

To configure the Upgrade Server via the Web configuration interface, open the web browser. Enter the GXP1450 IP address. Enter the admin password to access the web configuration interface. In the ADVANCED SETTINGS page, enter the Upgrade Server's IP address or FQDN in the "Firmware Server Path" field. Select TFTP or HTTP upgrade method. Update the change by clicking the "Update" button. "Reboot" or power cycle the phone to update the new firmware.

During this stage, the LCD will display the firmware file downloading process. Please do NOT disrupt or power down the unit. If a firmware upgrade fails for any reason (e.g., TFTP/HTTP server is not responding, there are no code image files available for upgrade, or checksum test fails, etc), the phone will stop the upgrading process and re-boot using the existing firmware/software.

Firmware upgrades take around 60 seconds in a controlled LAN or 5-10 minutes over the Internet. We recommend completing firmware upgrades in a controlled LAN environment whenever possible.

No Local TFTP/HTTP Server

For users who do not have a local TFTP/HTTP server, we provide a HTTP server on the public Internet for users to download the latest firmware upgrade automatically. Please check the Support/Download section of our website to obtain this HTTP server IP address: http://www.grandstream.com/support/firmware.

Alternatively, download and install a free TFTP or HTTP server to the LAN to perform firmware upgrades. A free Windows version TFTP server is available:
http://support.solarwinds.net/updates/New-customerFree.cfm.

INSTRUCTIONS FOR LOCAL TFTP UPGRADE:

1. Unzip the file and put all of them under the root directory of the TFTP server.
2. The PC running the TFTP server and the GXP1450 should be in the same LAN segment.
3. Go to File -> Configure -> Security to change the TFTP server's default setting from "Receive Only" to "Transmit Only" for the firmware upgrade.
4. Start the TFTP server, in the phone's web configuration page
5. Configure the Firmware Server Path with the IP address of the PC
6. Update the change and reboot the unit

User can also choose to download the free HTTP server from http://httpd.apache.org/ or use Microsoft IIS web server.

NOTE:

- When GXP1450 phone boots up, it will send TFTP or HTTP request to download configuration file "cfg000b82xxxxxx", where "000b82xxxxxx" is the MAC address of the GXP1450 phone. This file is for provisioning purpose. For normal TFTP or HTTP firmware upgrades, the following error messages in a TFTP or HTTP server log can be ignored: "*TFTP Error from [IP ADRESS] requesting cfg000b82023dd4 : File does not exist.Configuration File Download*"

CONFIGURATION FILE DOWNLOAD

The GXP1450 can be configured via Web Interface as well as via Configuration File (binary or XML) through TFTP or HTTP/HTTPS. The "Config Server Path" is the TFTP or HTTP server path for the configuration file. It needs to be set to a valid URL, either in FQDN or IP address format. The "Config Server Path" can be the same or different from the "Firmware Server Path".

A configuration parameter is associated with each particular field in the web configuration page. A parameter consists of a Capital letter **P** and 2 to 4 digit numeric numbers, i.e., P2 is associated with "Admin Password" in the ADVANCED SETTINGS page.

For a detailed parameter list, please refer to the link below to download the corresponding configuration template of the firmware.
http://www.grandstream.com/support/tools

Once the GXP1450 boots up (or re-booted), it will request a configuration file named "cfgxxxxxxxxxxx" followed by a request for configuration XML file named "cfgxxxxxxxxxxx.xml", where "xxxxxxxxxxx" is the MAC address of the device, i.e., "cfg000b820102ab". The configuration file name should be in lower cases.

For more details on XML provisioning, please refer to http://www.grandstream.com/support.

Managing Firmware and Configuration File Download

When "Automatic Upgrade" is set to "Yes", a Service Provider can use P193 (Auto Check Interval, in minutes, default and minimum is 60 minutes) to have the devices periodically check for upgrades at pre-scheduled time intervals. By defining different intervals in P193 for different devices, a Server Provider can manage and reduce the Firmware or Provisioning Server load at any given time.

Restore Factory Default Setting

WARNING: *Restoring the Factory Default Setting will **delete** all configuration information of the phone. Please **backup** or **print** all the settings before you restoring factory default settings. We are not responsible for restoring lost parameters and cannot connect your device to your VoIP service provider.*

INSTRUCTIONS FOR RESTORATION:

Step 1: Press "OK" button to bring up the keypad configuration menu, select "Config", press "OK" to enter submenu, select "Factory Reset" (Please refer to Table 5-1 of keypad flow chart)

Step 2: Enter the MAC address printed on the bottom of the sticker. Please use the following mapping:

0-9: 0-9
A: 22 (press the "2" key twice, "A" will show on the LCD)
B: 222
C: 2222
D: 33 (press the "3" key twice, "D" will show on the LCD)
E: 333
F: 3333

Example: if the MAC address is 000***b***8200***e***395, it should be key in as "000***222***8200***333***395".

NOTE: If there are digits like "22" in the MAC, you need to type "2" then press "->" right arrow key to move the cursor or wait for 4 seconds to continue to key in another "2".

Step 3: Press the "OK" button to move the cursor to "OK". Press "OK" button again to confirm. If the MAC address is correct, the phone will reboot. Otherwise, it will exit to previous keypad menu interface.